AETA AUDIO SYSTEMS

SCOOP STUDIO

User Manual

AETA AUDIO SYSTEMS S.A.S.
Parc technologique - Kepler 4
18-22, avenue Edouard Herriot - 92350 Le Plessis Robinson – FRANCE
Tél. +33 (0)1 41361200 – Fax +33 (0)1 41361269
Web : http://www.aeta-audio.com

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1. SCOOP STUDIO – Easy quick Start

1. Plug ISDN or POTS line to the appropriate socket on rear of unit
2. Plug in audio connections.
3. Power on the Scoop Studio via switch On/Off on rear panel of unit.
4. Select the appropriate network via the network menu
5.
   - To use direct dial “Number” mode , enter a number
   - To dial one of 5 last numbers press “green phone” key once.
   - To use a profile number, enter a letter

   For 2 last modes, select the profile number via keypad (left and right key)
6. Press “green phone” key again for dialling.
7. If busy or bad connection Press "red phone" then press the “green phone” key twice to redial the last number dialled.
8. Connection status will be displayed in LCD screen once connected.

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2. Introduction

2.1. Functions

The Scoop Studio is designed to enable radio broadcasters to conduct high quality live two-way remote broadcasts, or two way commentaries with return cue, via ISDN or POTS lines.[1]

2.1.1. Algorithms

The Scoop Studio contains a mono audio compressor/de-compressor (Codec) that performs all necessary ISDN and POTS algorithms.

In ISDN mode, the user can select one of four operational audio standards:

1. **Phone mode** (G.711, 3,5kHz)
2. **Live speech** (G.722, 7kHz, low delay)
3. **Music CD quality** (Layer II, 20kHz)[2]
4. **Live concert** (4SB-ADPCM,15kHz, proprietary low delay) [2]

In POTS mode, the user has only live speech mode (CELP , 7kHz)

One outstanding feature of Scoopy codec is the 5A System® on receiving an incoming ISDN call, the unit can automatically detect the coding algorithm and parameters of the calling codec, and then adjust itself in a compatible configuration so that the connection succeeds regardless of the initial configuration and that of the remote unit.

2.1.2. Audio interfaces

The Scoop Studio contains one audio input, one audio output and one headphone for monitoring

2.1.3. Transmission

Using an ISDN line, transmission bit-rate is either 64kbps or 128kbps[2]. Using a POTS line, transmission bit rate depends on the telecommunication network quality with a maximum bit-rate of 33.6 kbps. The Scoop Studio transmits data at a minimum rate of 12.000 bits and at a maximum of 24.000 bits of information a second

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[1] Depending on configuration version

[2] On the 15kHz ISDN version

® 5AS = Aeta Audio Advanced Automatic Adjustment System

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The Scoop Studio can work in many countries using various ISDN standards.

As ISDN protocol may vary from country to country, consult your AETA dealer before carrying your Scoop Studio abroad.

2.2. Applications

News remotes.
Live sport commentaries with local contributors.
Remote two-way interviews.
Remote contributions into studio discussions.
Live music concerts.

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3. Setting up the SCOOP STUDIO

3.1. Power

3.1.1. Optional DC supply

The Scoop Studio will also work on any external 8 to 15-volts DC source. A typical source will be a car cigarette adapter. Connect your DC power cord to the socket at the back panel of the unit (labeled DC In 8-15 V 2A), and plug the other end into your DC power source.

Note : See the chapter 14 for the connection.

3.2. Connection to the ISDN

Connect the (RJ45) connector of the ISDN cable into the socket on the back panel marked "ISDN", and connect the other end of the cable into the ISDN wall socket.

The ISDN modem of the scoop is a S/T or a U interface depending on the unit type.
You can select the correct ISDN protocol for a given country from the menu.

Given the various kinds of ISDN protocols used in different countries or inside PBXs, ISDN compatibility problems may occur. Please be sure to select the right protocol for the country you are in. In case oftrouble please contact your AETA dealer for advice.

3.3. Connection to the POTS

Connect the (RJ11) connector of the telephone cable into the socket on the back panel marked "ANALOG", and then connect the other end of the cable into the telephone wall socket.

The Scoop Studio's RJ11 connector will accept 4 or 6 conductor modular plug, but only the 2 center conductors, (typically Red & Green) are used.

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Caution: Every country has its own style of telephone connector. Consult your engineers, your local AETA dealer for further advice.

Dialing methods

Telephones dial numbers either by pulsing the line, (you will hear a "clicking" sound similar to that heard when dialing from a rotary dial telephone) or by sending audio tones (DTMF) The Scoop Studio can dial using either pulse or DTMF tones.

Caution:

Do not connect the Scoop Studio to a telephone jack that provides power for lighting a telephone's dial.
Do not connect the Scoop Studio to a party line or coin-operated telephone line.
Not suitable as an extension to a pay phone or use with a shared service line or 1+1 carrier system line.
You should disable call waiting if in use.

PBX and PABX applications

The internal modem of the Scoop Studio is only approved for use as an extension instrument to compatible PBXs.
Contact AETA AUDIO S.A. or your local dealer for an up-to-date list of PBXs with which the internal modem is compatible.
AETA AUDIO S.A can not guarantee that the Scoop Studio will operate correctly under all possible conditions of connections to compatible PBXs. Any cases of difficulty should be referred in the first instance to AETA AUDIO S.A.

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4. SCOOP STUDIO STRUCTURE

4.1. Front view

Figure 1 - Front panel

1 - LCD display
2 - Function keys
3 - Status LED's
4 – Keypad
5 – Bargraph
6 – Monitoring selection
7 – Headphone socket
8 - Headphone volume adjustment

4.2. Scoop Studio status

There are 28 LED's on the front panel providing the following information :

- Info (2 yellow LED's) : Relay status
- Alarm (red) : When "on", indicates a network problem.
- Dec (green) : When "on" indicates that a successful connection exists and the Scoop Studio is decoding the POTS or ISDN signal.
- Tx Level meter : 11 LED's (scale –20 to + 5 VU) + an Overload led
- Rx Level meter : 11 LED's (scale –20 to + 5 VU) + an Overload led

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4.3. General synoptic diagram

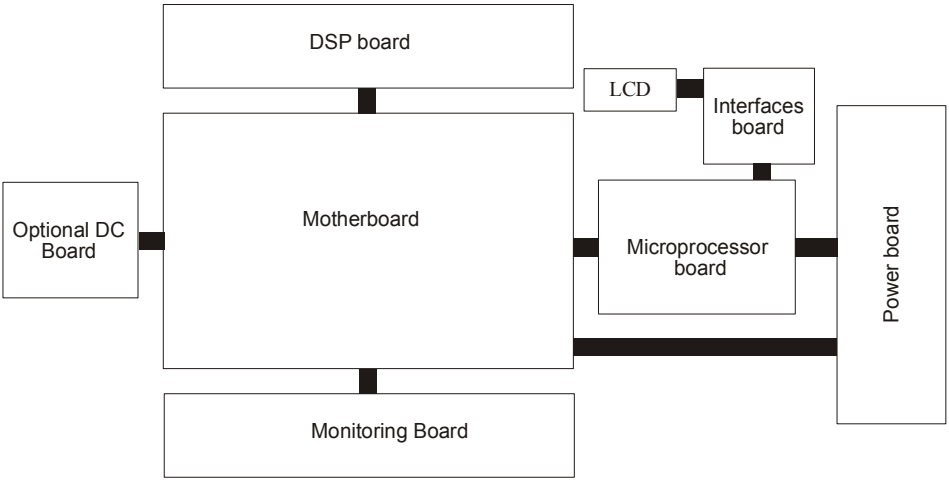

Figure 2 – Scoop Studio synoptic

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5. Audio section

5.1. Encoding and decoding

Scoop Studio use includes a wide range of coding algorithms. First, one can select among algorithms compliant with ISO and ITU-T[1] recommendations :

- G711;
- ITU-T G722 (mono at 64 kbit/s);
- MPEG Audio Layer II at 48, 32, 24 or 16 kHz, with programmable channel mode and bit rate ;

MPEG Audio and G722 algorithms comply with ITU-T J52 recommendation for ISDN transmission. Besides, other algorithms are available, that are so-called “proprietary” because they do not comply with enforced standards :

- Proprietary MPEG Layer II at 64 kbit/s or 128 kbit/s (for compatibility with ISDN codecs that are not compliant with the J52 recommendation) ;
- 4SB ADPCM, running in mono at a 128 kbit/s bit rate; the bandwidth with this algorithm is 15 kHz ;
- TDAC mono, running at 64 kbit/s, with a 15 kHz bandwidth ; available as an option.

The following describes some important features of the various available algorithms and protocols.

5.1.1. Notes about G711

G711 is the standard coding used for voice transmission on public telephone networks. This algorithm is used for links (via ISDN) with telephones or hybrid devices.

[1] former CCITT

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5.1.2. Notes about G722

With G722 coding, two synchronisation modes are available:

- "Statistical recovery" byte synchronisation method (alias SRT) ;
- H221 synchronisation; in this case, 1.6 kbit/s from the compressed data are used for this.
- H221 synchronisation and H242 protocol

H221 synchronisation is highly recommend when possible, as it features higher reliability and faster recovery time, while degradation (because of the bit rate used for framing) is minimal.

H242 synchronisation is recommended by ITU-T, and is included in J52. However, the mode with H221 synchronisation but without H242 protocol can be useful for compatibility with old generation codecs which did not use this protocol.

5.1.3. Notes about J52 and MPEG coding

The ITU-T J52 recommendation was defined in order to allow the interoperability of various equipment over the ISDN[1], using common coding standards. It includes the following features:

- Interoperation procedures as per ITU-T H242 recommendation ;
- In the case of MPEG encoding, optional protection against transmission errors (Reed-Solomon error correction codes).

It must be noted that, thanks to the interoperation protocol, J52 codecs, when setting up a link, can negotiate automatically and agree on a configuration that is compatible with the capability of both units (regarding bit rate, channel mode, etc.). In this way, when the units differ in their capability (or make), the resulting configuration may be different from expected beforehand, but in most cases the link will work and audio will be transmitted.
As another useful consequence, this also gives users more tolerance to mistakes when configuring the units on the two sides of the transmission links, as the codecs will adapt automatically even with differences in the initial settings of the two units.

[1] J52 is not needed nor applicable to leased line connections

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5.1.4. Notes about TDAC

As an option, the codec can also include the TDAC algorithm. TDAC is for Time Domain Aliasing Cancellation ; this is a transform coding based on an MDCT (Modified Discrete Cosine Transform), encoding a 15 kHz bandwidth mono signal at a 64 kbit/s bit rate. When the option is installed, three modes are available :

- TDAC mono full-duplex, running at 64 kbit/s, with a 15 kHz bandwidth ;
- G722/TDAC : G722 encoding, TDAC decoding, running both in mono at 64 kbit/s ;
- TDAC/G722 : TDAC encoding, G722 decoding (with SRT), running both in mono at 64 kbit/s ; this mode is symmetric to the previous one.

5.1.5. Symmetric or asymmetric codec modes

The codec allows two communication modes:

Symmetric communication: in this mode, the encoder and decoder both use the same coding algorithm with the same configuration (channel mode, etc.). In this case, the communication is strictly symmetric full-duplex, with exactly the same coding configuration used in both directions (local to remote and remote to local). This is usually required when using proprietary algorithms.

Asymmetric communication: this mode is used for applications requiring different coding configurations in the two directions. The J52 protocol allows such mode. To give some examples, it is possible to transmit MPEG in one direction and G722 in the other one.

With the TDAC option, asymmetric modes are also available wherein one direction is G722 coded while the other one is TDAC coded. Such mode is useful e.g. in order to get a low delay return path encoded in G722 while the send path is encoded with higher quality but a higher delay.

5.1.6. Relays

In ISDN mode with MPEG L2 J52 or 4SB ADPCM and in PSTN mode (Secured mode must be disable) you have access to the relays.

- You can activate remote relays by pressing key “F1” for the first relay and “F3” for the second relay : Option “Relays(Keys)”, “On”

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- You can do the same thing with the 2 optos : Option “Relays(Optos) ”, “On”
- When you press one of these keys or enable an opto input :
  - On the LCD of the local unit, we display “C1” or “C3” “ON” depend on the key pressed.
  - On the remote unite, we activate relay(s) 1 or/and 2 and the LED “INFO 1” or/and “INFO 2” light(s) on .

Note :

In PSTN mode you have only access to the first relay.

5.2. Audio Interfaces

5.2.1. Analogue audio Input

Audio characteristic are measured over a 20 to 20kHz bandwidth except when diffently stated.

Format	balanced
Connector	3-pin female XLR socket
Maximum input level	+0 to +22 dBu by step pf 1dB (menu)
Input impedance	10 kΩ
Common mode rejection ratio	>60dB @ 1kHz

Table 1 – Input interface

5.2.2. analogue audio Outputs

The audio signal output is available on the line level output.

The monitoring source can be changed with the headphone key.

- If the led in the extension of the Tx bargraph is light, you have the local audio signal on the headphone.
- If the led in the extension of the Rx bargraph is light, you have the return audio signal on the headphone.

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- If the same both leds are light, you have the local audio signal on the left ear and the return on the right ear.

Line Out Interface :

Format	Balanced
Connector	3-pin male XLR socket
Maximum output level	+0dBu to +22dBu by steps of 1dB (software selection)
Output impedance	$\leq 50\ \Omega$
Output symmetry	> 60 dB

Headphone Interface :

Connector	6.35mm jack socket
Maximum output level	+20dBu
Load impedance	$\geq 16\ \Omega$

5.3. Audio performance

A) Analog performance

Measurement condition:

- AD/DA Loop
- Sample frequency : 48kHz

Maximum Gain (Input to Output)	+22dB
Signal to Noise ratio	84dBrms
Bandwidth	20Hz – 20 000 Hz ± 0.5dB
Distortion (THD+N)	< 80 dB (0.01%) @ 950Hz

Table 2 – Audio performance

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B) In ISDN mode

Data rate	Sample frequency	Bandwidth	Delay	Algorithm
128 kbit/s	48kHz	20Hz - 20kHz	137ms	MPEG II (- / J52)
128 kbit/s	32kHz	20Hz – 15kHz	202ms	MPEG II (- / J52)
128 kbit/s	24kHz	20Hz – 10.4 kHz	268ms	MPEG II (- / J52)
128 kbit/s	16kHz	20Hz – 7.2 kHz	398ms	MPEG II (- / J52)
128 kbit/s	32kHz	20Hz – 15 kHz	7ms	4SB ADPCM
64 kbit/s	48kHz	20Hz – 20kHz	163ms	MPEG II (- / J52)
64 kbit/s	32kHz	20Hz – 13.4kHz	202ms	MPEG II (- / J52)
64 kbit/s	24kHz	20Hz – 10.4 kHz	268ms	MPEG II (- / J52)
64 kbit/s	16kHz	20Hz – 7.2 kHz	400ms	MPEG II (- / J52)
64 kbit/s	32kHZ	20Hz – 15kHz	80ms	TDAC
64 kbit/s	16kHz	20Hz - 7kHz	11ms	G722 SRT/H242
64 kbit/s	16kHz	300Hz – 3.5kHz	17ms	G711- phone

Table 3 – ISDN mode

Note : In MPEG II without J52, Scoop Studio is compatible with other manufacturer codecs.

C) POTS mode - CELP Algorithm

Data rate	Audio quality
12Kbit/s	3.6kHz
14.4Kbit/s	4.3kHz
16.8Kbit/s	5.1 kHz
19.2 Kbit/s	5.7 kHz
21.6 Kbit/s	6.3 kHz
24.0 Kbit/s	**7.2 kHz**

Bandwidth : 40 Hz to 7 kHz (@ 24 kbps data rate)

24 kbit/s can typically be achieved in all countries that support V.34 modems on their public switched networks.
Higher rate depends on line quality.
The CELP algorithm is optimised running at 24 kbit/s.

Table 4 – CELP

Note : CELP is a proprietary algorithm of France Telecom CNET

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5.4. Audio monitoring

The 12 LED's labeled “Tx” on the front side of the Scoop Studio indicate the peak level of the sending audio signal**.** The level display reference (0 dB) is 8 dB below the clipping level.

The 12 LED's labeled “Rx” on the front side of the Scoop Studio indicate the peak level of the receiving audio signal**.** The level display reference (0 dB) is 8 dB below the clipping level.

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6. SCOOP STUDIO OPERATION: How it works.

6.1. Introduction

Incoming audio into the Scoop Studio is digitized by a state-of-the-art A/D converter and processed through the Scoop Studio's codec. The data is then sent via the internal ISDN or POTS synchronous modem to the telephone network (ISDN or POTS) to a remote Scoop Studio or another compatible ISDN Audio Codec. Operating with a very fast DSP, the codec runs an algorithm modeling the digital audio signal, in order to reduce the digitized audio data rate.
At the other end of the telephone network, the answering Scoop Studio reconstructs the original audio signal with very little loss or induced artefacts and at an extremely low audio delay time.

6.2. User interface

The user interface consists of a lexan matrix keypad and a LCD display. The keypad has two sections.

- The first section is a 4x3 matrix including the numbers from 0 to 9,"*", "#".

  Some keys have many functions :
  2, 3, 4, 5, 6, 7, 8, 9, 0: for accessing to letters display on the key, press the key several times.
  Note : Space character is available on the "1" key.

- The second section is the Extended Keypad functions under the display.
  There are 3 function keys not labeled. The key function depends of the menu, the function label appears over the key on the second line of the display

- The third section is the special Keypad functions.
  "OK" key to validate a choice.
  "Esc" key to escape from a menu.
  "Green phone" key to make a call.
  "Red phone" key to on hook a call.

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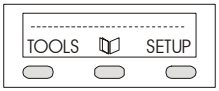
6.3. Scoop Studio Menu

1) Main menu



To scroll in the sub-menus use the keys under the word
If a second sub-menu exists, you can enter by pressing again the key under the word
At any time you can return to the main menu by pressing the Esc key.

Note : the symbol between "TOOLS" and "SETUP" means : "DIRECTORY"

If you press "green phone" key, you access to the 5 last called number.
If you enter a letter, you access to the remote directory.
If you enter a number, you can make a direct call.

Note : If you have a restricted menu, you can disable it by pressing the directory key and this following sequence: "1","6","4","3"

6.3.1. Scoop Studio default configuration

The Scoop Studio's "General reset" set default configuration is useful to configure the modem in case communication difficulties are encountered.

Note : The stored calling numbers are not erased when you make a General reset.

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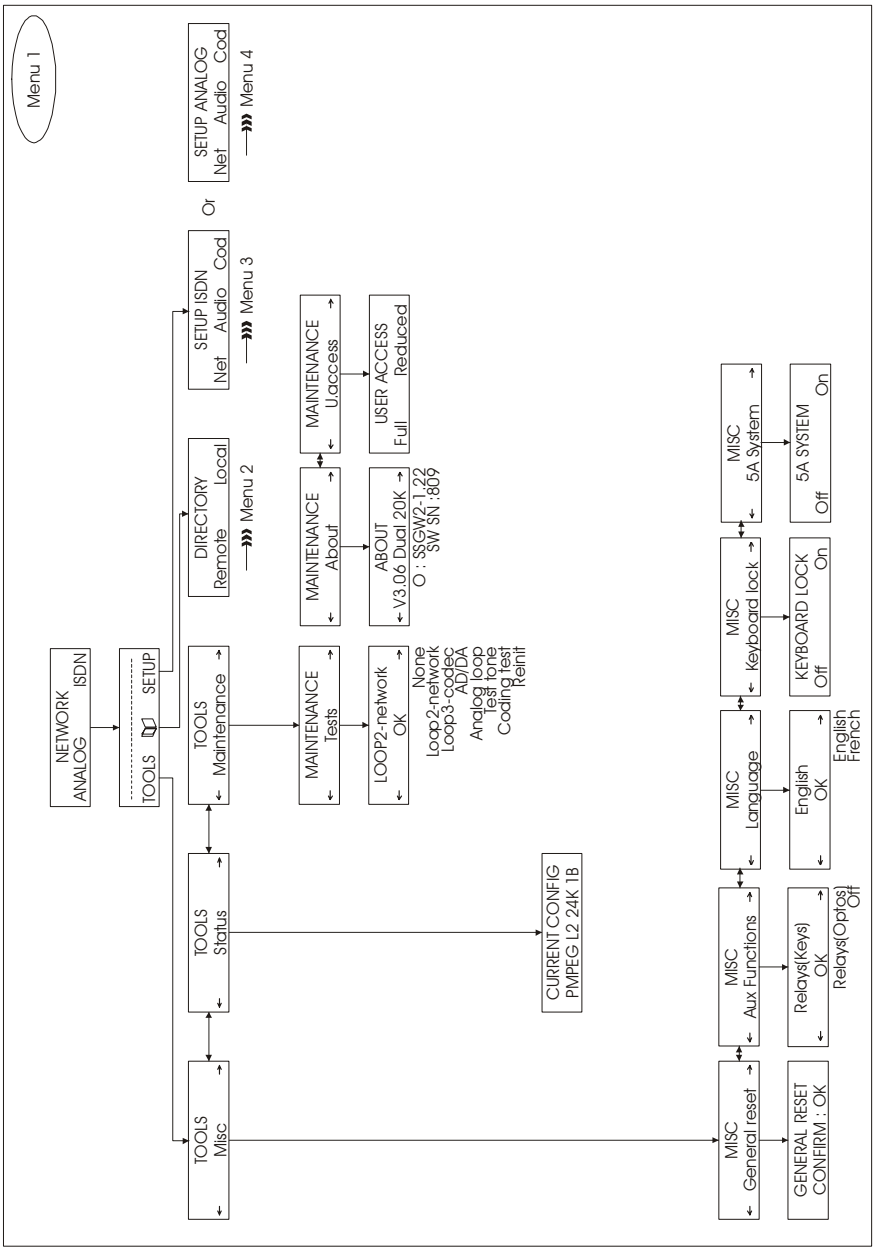

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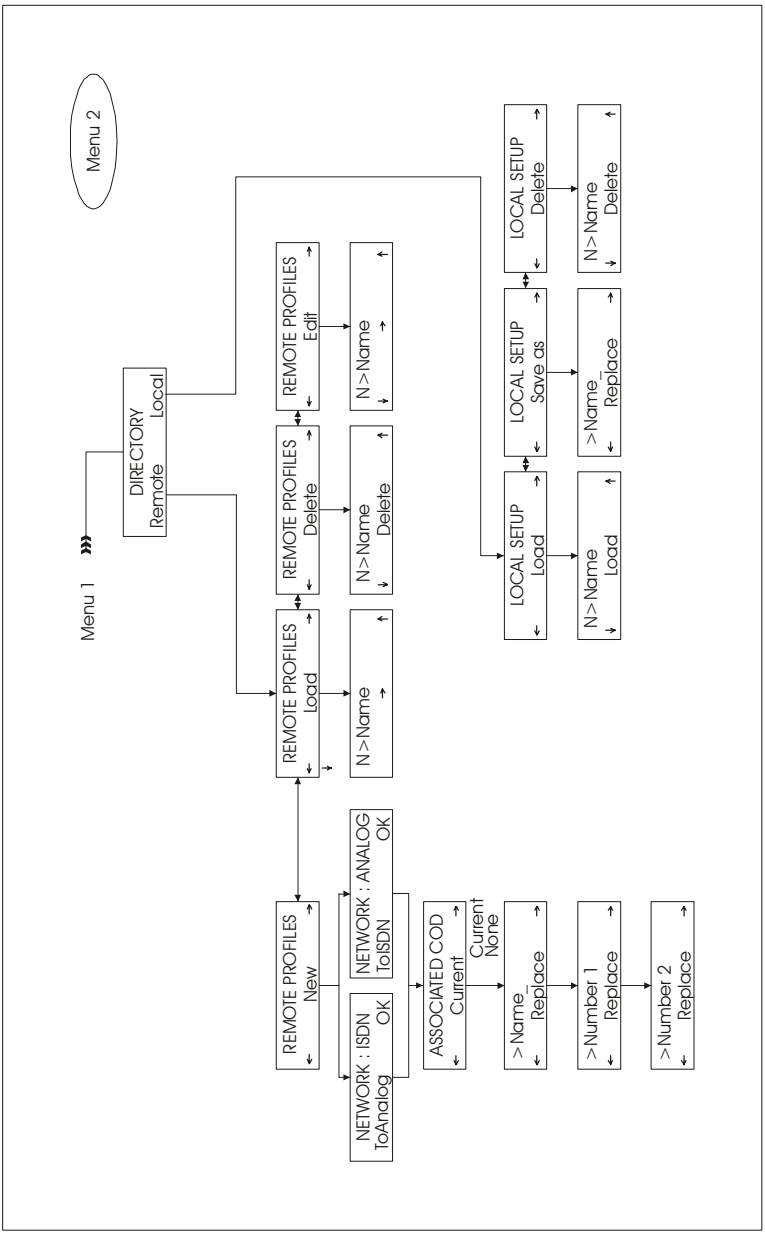

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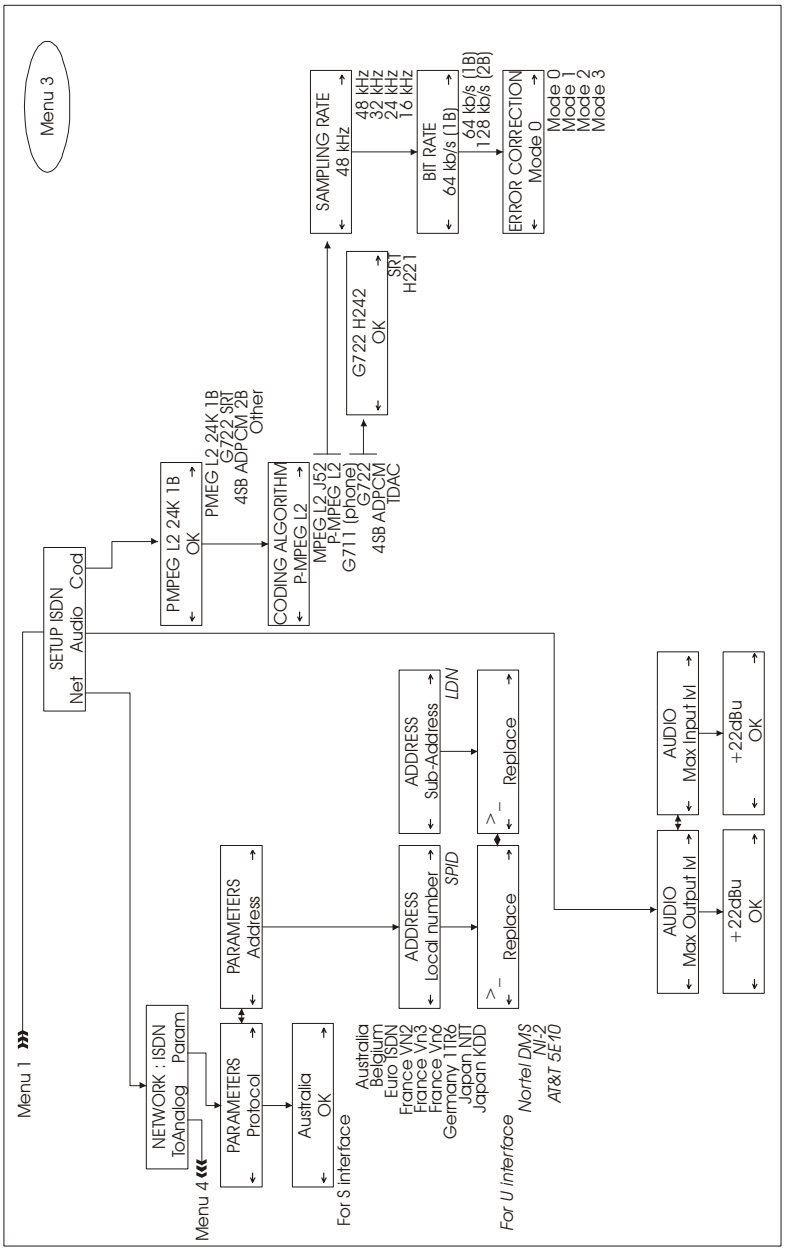

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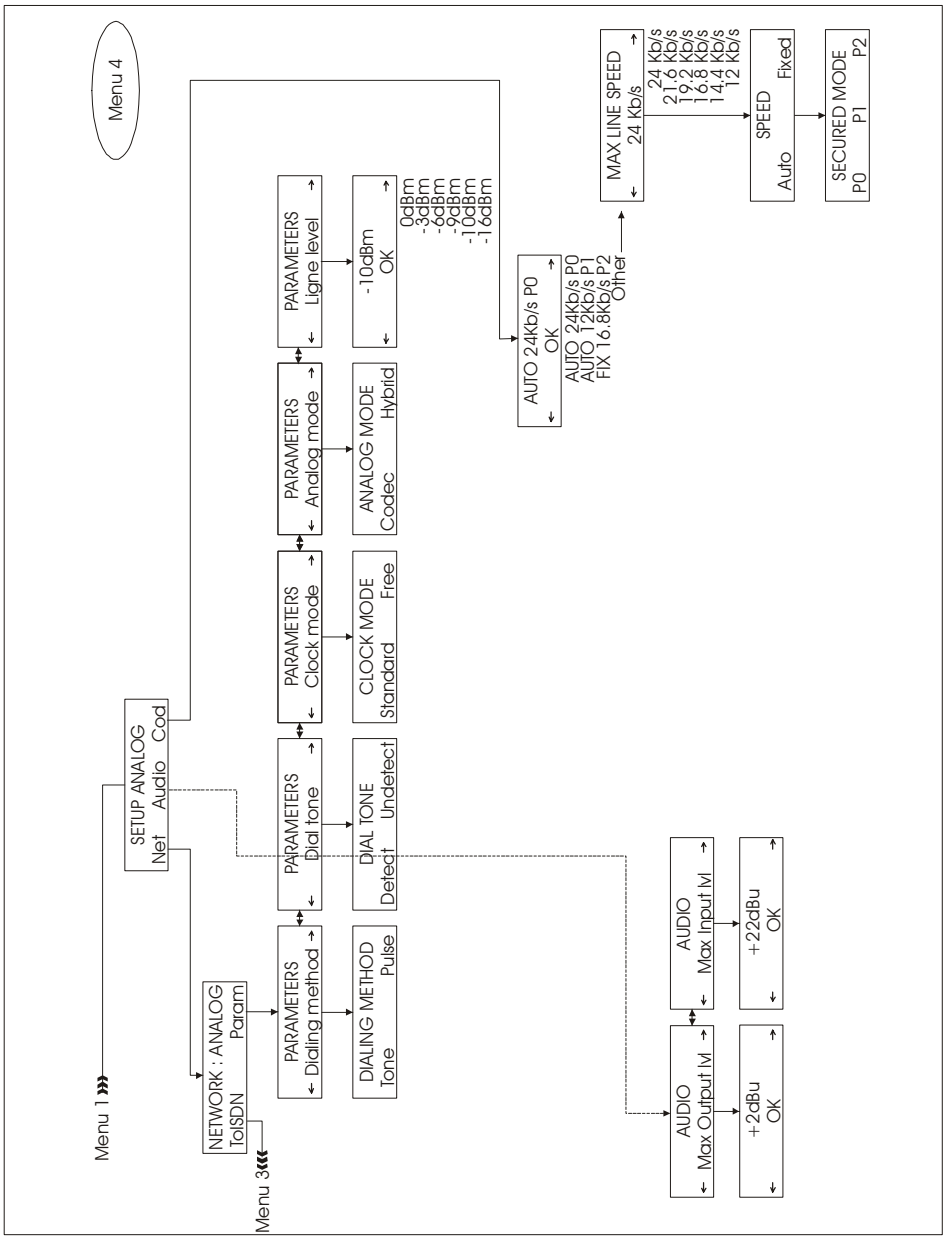

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7. How to Set-Up Profiles on the Scoop Studio

You can access and edit memory locations and Profiles directly from the front panel keypad and menus of the Scoop Studio. The Scoop Studio has the ability to allow you to pre-program the 100 Profile locations.

7.1. What is a profile ?

A profile is a non-volatile, pre-programmed memory location stored within the Scoop Studio which functions very similar to the auto dial memory locations on an average telephone.

A remote profile can contain an ISDN or POTS number with specific parameters associated with that number. The remote profile can contain the name of the location to be dialled and its specific algorithm. You can create up to 50 unique remote profiles on the Scoop Studio. If the remote profile is an ISDN type, you can have two numbers stored (one number for each B channel).

A local profile (named local setup for making the difference) can contain ISDN or POTS local parameters. The local setup can contain the name and all network parameters like local address in ISDN. You can create up to 50 local setups on the Scoop Studio

7.2. How to manage profiles on the Scoop Studio

Note: All remote profile can be a POTS profile or an ISDN profile.

7.2.1. How to manage remote Profiles

From the Main Function Menu, select "DIRECTORY" symbol, and press the key under it. After having selected "REMOTE" choice, you can select different actions :

- "Load" for loading in memory a profile.

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- “Delete” for deleting a profile.
- “Edit” for looking each elements of the profile.
- “New” for creating a new profile.

“Load”

With the left and right extended display key, you can scroll the remote profiles. If you enter a letter on the keypad, the profile list scroll to the profile whose the name begin with this letter.

With the centre extended display key, you scroll each profile items.

For calling with the display profile, you have just to press the “green phone” key.

For loading in memory the profiles, press the “OK” key.

“Delete”

With the left and right extended display key, you can scroll the remote profiles. If you enter a letter on the keypad, the profile list scroll to the profile whose the name begin with this letter.

With the centre extended display key, you delete the display profile

“Edit”

With the left and right extended display key, you can scroll the remote profiles. If you enter a letter on the keypad, the profile list scroll to the profile whose the name begin with this letter.

With the centre extended display key, you scroll each profile items.

Press “OK” key to modify one or many items of the display profile.

“New”

At first you should select the network : Analogue/POTS or ISDN.

After that, you have the choice to associate an algorithm (current algorithm configuration) or not to your profile. If you don’t associate an algorithm to your profile,

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when you will make a call with this profile, Scoop Studio will use the current algorithm configuration.

The next stage is to enter a name for your profile, and press “OK” to valid.

Now, Scoop Studio ask the user to enter one or two numbers (depend of algorithm configuration). If you don’t enter numbers, when you will make a call with this profile, Scoop Studio will ask you to enter the numbers.

7.2.2. How to manage local setups

From the Main Function Menu, select “DIRECTORY” symbol, and press the key under it. After having selected “LOCAL” choice, you can select different actions :

- “Load” for loading in memory a local setup.
- “Save as” for creating a new local setup.
- “Delete” for deleting a local setup.

In a local setup memory we save the current network configuration.

For example : If you are in ISDN, we save the ISDN protocol, the local address and the local sub-address.

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8. Connecting 2 SCOOP STUDIO.

Note: The following is valid for both POTS and ISDN mode.

Warning: In ISDN mode with some PBX's, you must enter your local number and your SPID number prior making a call.

8.1. Initiating a call

There are 3 ways to initiate a call :

- Dialing with a profile
- Direct Dialing
- Re-dialing the previous number.

8.1.1. Dialling Using a Profile Number

From the Main Function Menu, select "DIRECTORY" symbol, and press the key under it. After having selected "REMOTE" choice, press the key under "Load".
Note : You arrive directly in the remote profile list, when you enter a letter under the main menu.

Select the profile number and press "Green phone" key.

"Call XXXXXXXX" appears on the screen and is dialed automatically.

8.1.2. Direct Dialling

Enter the telephone number and press the "green phone" key. If you call two numbers (In ISDN), you need to press again the same key.

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Note : In the case that you have 2 numbers, if you don't enter the second number, we call twice the first number.

A message to indicate that the call is in progress is displayed on the screen.

***Note** :*

- The number length is limited to 23 digits and may be displayed on 2 lines.

- Insert a "∗" between number and sub-number in ISDN mode.

- Insert a "∗" for wait in POTS mode

8.1.3. Re-Dialling the Previous Number

From the main menu, press "green phone" key, The last called number appears on the screen, press again the "green phone" key. Actually, your are in a short list of the five last called numbers.

"Call in progress" along with the redialed number is displayed on the screen.

Note : We don't re-load configuration, we use the last configuration used (current now).

Note :
In case of mistake you may come back at the beginning of the menu by pressing the "Esc" key.

Note :
As soon as the local and remote Scoop Studio are connected, the CONNECT result code is displayed.
If a connection can not be established, the NO CARRIER result code will be displayed.
The bit rate is displayed in POTS mode.

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8.2. Disconnecting a call

To end a call press the "red phone" key.
"Wait..." is displayed, after awhile, the Scoop Studio is reset and ready for the next call.
The main menu appears on the screen.

8.3. Auto Answering

As soon as the "Power On Initialization" phase is completed, the Scoop Studio is ready to receive an ISDN call or a POTS call. You have just to adjust the levels in such a way that the audio level green and yellow LED's indicate a normal operating range.

When a call is received the Scoop Studio will recognize ISDN or POTS and establish a connection. Adjust your headphone level and your local feed back with the local/return mix balance if needed.

Then the Scoop Studio is ready for full duplex audio communication.

8.4. Entering local Numbers

From the Main Function Menu, go to “SETUP”, “NET”,”PARAM”. Using the down arrow key scroll right to "Address" screen. Press the center key to enter in address configuration

You have two address configuration with each sub-address. You have a specific address and sub-address for each B ISDN channel.

A series of AT commands will be displayed and automatically return you to the main menu

***Note** : In many cases, the sub-address is not necessary.*

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8.5. Entering SPID Numbers (USA)

In the USA, some ISDN circuits require two SPID numbers and two LDN (Local directory number), one SPID for each B channel, in addition to the local dialing number. The Scoop Studio can be manually programmed using the keypad.

From the Main Function Menu, go to “SETUP”, “NET”,”PARAM”. Using the down arrow key scroll right to "Address" screen. Press the center key to enter in address configuration

Note :

- You should enter SPID 1 and LDN 1 local number and before SPID 2 number and LDN 2
- Generally, the LDN is the 4 last digits of the SPID number.

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9. POTS Information

9.1. Factory default configuration

The Scoop's factory-set default configuration is suitable for most Scoop Studio transmission applications and are reloaded by the selected function :
“TOOLS”, “Misc”, ”General reset”.

Your Scoop Studio is designed to operate over dial-up phone circuits with the following dialling and call monitor features :

- Multi-frequency signalling (Tone dialling method) or
  Loop-disconnect signalling (Pulse dialling method)

Rem :
By issuing the configuration procedure the user can change the <dialling method> : Pulse or Tone
The selection of the dialling method will be stored until the user has to modify his choice again even when the Scoop Studio is power off.

- Operation in the absence of proceed operation (waiting for dial tone)
- Automatic answering
- Originating and answering handshake negotiations begin at the highest DCE line speed configured in the factory (24 kbps) or at lower speed selected by the menu with the current configuration.
  Automatic speed selection : Handshake negotiations fall back to a lower speed if necessary.
- Full dial progress detection (Dial tone detect).

Rem :This parameter must be "disable" for calls originated from Switzerland and Italy.

Additional setting

- Fall back if negotiation fails at the highest speed (speed automatic)
- Maximum DCE Line speed = 24000 bps. (Default Config.)

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9.2. POTS modes

9.2.1. Phone / Codec Mode

The telephone mode or the codec mode is selected by the menu “ANALOG MODE”. If you select Hybrid parameter, you select the phone mode.

The change of the initial mode can’t be done during the communication.

When the high speed connection is broken the Scoop Studio doesn't fall back to the regular POTS mode (3.4 kHz). The voice communication is interrupted and a new call has to be made.

9.3. Network parameters

9.3.1. Setting the optimal rate

The Scoop Studio with the lowest max line speed setting will determine the maximum connect rate.

a) When the speed mode function is set on "Automatic" adaptation (general reset Configuration.) both modems will negotiate the highest transmission rate according to the quality of their current respective networks.

   This rate is also limited at the lowest speed of the two max speed selected on the 2 units.

   If the line quality is changing during the audio-transmission the modems will try to adapt consequently the data rate by fall back at a lower data rate and fall forward to the higher selected speed. During each re-negotiation the audio signal may be interrupted. If these "break down" appear, it is highly recommended to set the max line speed selection of one of the Scoop Studio at one level or two below the used connect rate.

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b) When the speed function is set on "Fixed" at ONE of the both Scoop Studio unit, the 2 modems will be allowed to negotiate at only the lowest speed of the two max speeds selected.

They will neither "fall forward" nor "fall back".

If this select speed is too high for the possibility of one of the local network capacity, the modem will "NOT CONNECT" and a lower has to be selected by the user to obtain a solid connection at a reliable data rate.

Rem : By setting Speed fixed, the user has the ability to select the max/min connect rate for the modems before a call is placed. So the fall back to a lower speed cannot occur during the communication.

9.3.2. Clock Mode

For long distance transmission, it is better to set on free this parameter. The reason is : in this mode, each modem generates the transmit clock and generate receive clock from receive carrier signal. In this case, each way is separated.

In standard mode, each modem works with only one clock. The local Scoop Studio generate the clock and the remote generate its clock from receive carrier signal.

9.3.3. Line level

The phone line level depending of country where you are. You can adjust the phone line level between 0dBm to –16dBm. The most popular level is –10dBm (default value).

9.4. Error protection

This function reduces the short and occasional transmission errors causing glitches and drop-out in the audio. Those errors can be founded particularly on long distance circuits and when connected to an in-house phone systems.

The audio quality could be slightly affected.

The user will only hear a short additional delay.

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9.4.1. Secured modes

Three protection modes are available :

Protected 0 (Unprotected) is set by default (Factory Configuration)

- Is compatible with all Scoop Studio units.
- Keep the smallest transmission delay (coding : decoding) of 80 ms.

Protected 1

Ought to be selected manually and recovers errors of 100 ms.

Protected 2

Ought to be selected manually and recovers errors of 250 ms.

Delay :

Protection \ Data rate	24K	21.6K	19.2K	16.8K	14.4K	12K	Coding Test
P0	88ms	90ms	105ms	114ms	131ms	138ms	55.5ms
P1	293ms	320ms	348ms	394ms	456ms	535ms	255ms
P2	581ms	647ms	723ms	823ms	961ms	1146ms	553ms

Note : If errors still exist in protected mode 2, set the max line speed at the next lower speed.

Caution : **The same protection level configuration must be selected at the both end units.**

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10. Troubleshooting

Network Indication:

- Alarm (red)

  When “on” indicates a network problem.
  Check your network.

- Dec (green)

  When "on" indicates that the signal is decoded by the Scoop Studio.

Unable to establish a connection:

Check the RJ connection between the Scoop Studio and the telephone network. (RJ 11, identified as Tel on the rear panel of Scoop Studio for POTS, and RJ45, identified as ISDN on the rear panel of the Scoop Studio for ISDN)

Connection In ISDN mode

To test your ISDN line, you may connect an ISDN phone or other suitable ISDN verification device into the RJ45 connector instead of the Scoop Studio and call an ISDN number to verify a working ISDN line. Check the ISDN protocol, check the number, and check appropriate setting if going through a PBX.

Connection In POTS mode

To test your POTS line, you may connect a normal phone to the wall connector instead of the Scoop Studio unit and call a normal phone number. Check for proper POTS line settings: Dialing method, " Pulse/Tone", dial tone "Detect/Undetected". Check proper setting if going through a PBX (you may need to dial to get an outside line, Ex 9*).

If the Scoop Studio disconnects while on-line, check for loose connections between the Scoop Studio and the telephone connection. Line noise or interference may be interfering with the modem signals. Retry the connection by dialing the number again.

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11. Tests

11.1. Audio section testing

1- Analogue section test

a) Connect an audio signal to one of the audio inputs. That signal is available on the headphones (Potentiometer turns that feedback feature off when fully counter-clockwise).

b) Select the menu **< TOOLS > , < MAINTENANCE > , < Test >** : **AD/DA Loop**.

The test is OK if you get the audio signal either on the headphones, or Aux. Out.

To end the test go back to the test menu, disable the **AD/DA Loop** by pressing the “none” choice (The star appears on None configuration).

2- Digital and analogue parts
The encoder may be connected to the decoder locally to test digital circuits.

Connect and select an audio signal to any of the inputs.
Select the menu **< TOOLS > , < MAINTENANCE > , < Test >** : **Coding test**.

The message "Coding process" appears on the screen.
The test is OK if you get the audio signal either on the headphones, or Aux. Out.
To end the test press the "**Esc**" key, wait signal “……” is displayed on the screen, then the unit returns to standby**.**

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11.2. Network test

These allows checking the network and the remote codec.

1- The unit can be configured to loop back to the network the received data.

Select the main menu **< TOOLS > , < MAINTENANCE > , < Test >** : **Loop 2 - Network**.

The loop is enabled as soon as the unit is connected.

2- The unit can be configured to loop back to the network the sent data.

Select the main menu **< TOOLS > , < MAINTENANCE > , < Test >** : **Loop 3 - codec**.

The loop is enabled as soon as the unit is connected.

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12. ISDN modem information

12.1. ISDN Protocols

ISDN modem supports worldwide ISDN signaling (CCITT I.430, Q.921, Q.931) for voice/audio and data including the following network operator variants :

With USA software

- AT&T 5^E5, 5^E9, 5^E10
- Northern Telecom (DMS-100),
- National ISDN-1 and 2 (North America),

With other countries software

- France Telecom EuroNumeris (Vnx) with supplementary services,
- Deutsche Telekom 1TR6 and EuroISDN,
- NTT INS-64 (Japan),
- KDD ISDN (Japan),
- Telecom Australia Austel TS-013,
- All EuroISDN carriers (Austria, Denmark, Holland, Ireland, Italy, Norway, Portugal, Spain, Switzerland, United Kingdom,...).

12.2. ISDN CLEARING CAUSES

The following table lists the call clearing causes (returned for example in a **CLEARED:** message). Call clearing cause is in hexadecimal. Message meaning are given for an **ETSI ISDN**. Causes with values greater than 80 hex are generated internally.

Example : CLEARED 18: 12 means for the 18 reference system call there are no user responding.

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01	unallocated (unassigned) number
02	no route to specified transit network
03	no route to destination
06	channel unacceptable
07	call awarded and being delivered in an established channel
10	normal call clearing
11	user busy
12	no user responding
13	no answer from user (user alerted)
15	call rejected
16	number changed
1A	non-selected user clearing
1B	destination out of order
1C	invalid number format
1D	facility rejected
1E	response to STATUS ENQUIRY
1F	normal, unspecified
22	no circuit/channel available
26	network out of order
29	temporary failure
2A	switching equipment congestion
2B	access information discarded
2C	requested circuit/channel not available
2F	resources unavailable, unspecified
31	quality of service unavailable
32	requested facility not subscribed
39	bearer capability not authorized
3A	bearer capability not presently available
3F	service or option not available, unspecified

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41	bearer capability not implemented
42	channel type not implemented
45	requested facility not implemented
46	only restricted digital information bearer capability is available
4F	service or option not implemented, unspecified
51	invalid call reference value
52	identified channel does not exist
53	a suspended call exists, but this call identity does not
54	call identity in use
55	no call suspended
56	call having the requested call identity has been cleared
58	incompatible destination
5B	invalid transit network selection
5F	invalid message, unspecified
60	mandatory information element is missing
61	message type non-existent or not implemented
62	message not compatible with call state or message type non-existent
or not	or not implemented
63	information element non-existent or not implemented
64	invalid information element contents
65	message not compatible with call state
66	recovery on timer expiry
6F	protocol error, unspecified
7F	interworking, unspecified
91	no signaling data link establishment
A2	no line activation
FF	call clearing, unspecified

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13. How to open a SCOOP STUDIO ready for servicing

Tools required:

Philips Screw Driver (medium-sized)

The Scoop Studio will be now separated into three Sections :

1) The top metal panel
2) The casing lid

Steps :

- Switch to off Scoop Studio
- Remove the external mains power cable if you use it
- Undo screws at the top
- Remove the top panel

Now, you have access to all internal boards.

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14. Connectors layout

14.1. Remote Connector

This interface uses a 9-pin female Sub-D connector on the rear panel. This is a V24/RS-232 type interface with only Tx and Rx signals (no flow control). The following table indicates its pin out (DCE type pin out)

Pin	Description
2	TX , To the PC
3	RX, From the PC
7	CTS, From the PC
8	RTS , To the PC
5	Signal ground

14.2. AUX Connector

Pin		**Name**	Description	Function and remarks	Direction
1		DGND	Ground		
	9	STOR1-	Relay 1	Relay contact 1	O
2		STOR1+		Corresponds to “INFO 1”	
	10	STOR2-	Relay 2	Relay contact 2	O
3		STOR2+		Corresponds to “INFO 2”	
	11	STOR3-	Relay 3	Decoder status;	O
4		STOR3+		follows the status of the “DEC” LED.	
	12	N.C.			
5		+5V	Power		
	13	ETOR1-	Opto 1	Opto input 1	I
6		ETOR1+		choose resistor to allow for 10 to 20mA	
	14	ETOR2-	Opto 2	Opto input 2	I
7		ETOR2+		choose resistor to allow for 10 to 20mA	
	15	ETOR3-	Opto 3	Not used	I
8		ETOR3+			

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Remarks:

- Direction: I = input, O = output
- Relay/Opto 1 and relay / Opto 2 are the transparent contacts (relay transmission)
- Relays and opto inputs are bi-directional, i.e. the polarity does not care

Input loop control current : 6mA (max 100mA)
Resistance of input loop : 470Ohms (current limiting series resistor)
Maximum switching voltage (output) : 400V peak
Maximum switching current (output) : 250mA
Resistance of output loop : 8 Ohms Max

A +5V power supply is available on the connector, with 100mA capacity, witch can be used e.g. to power the input loops.
A +5V or +12V source may be connected directly on an input loop, because the internal series resistor is dimensioned for this purpose. For a higher voltage source, it may be necessary to limit the input current.

14.3. POTS Interface

The telephone network connection on Scoop Studio is a RJ 11 connector. (Labeled Analog)

Connector :

Pin	Description
1	-
2	TIP
3	RING
4	-

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Impedance adaptation (Internal rear panel switches):

Country	RZDC	ZZNT
US	8.2 Ω	600 Ω
CTR-21	22.1 Ω	600 Ω//0.0047µF

Country \ Switch	1	2
US	ON	OFF
CTR-21	OFF	ON

14.4. ISDN Interface Network

The ISDN connector is an RJ45 – 4 wires into 8 wires.

S0/T0 Network:

Pin	Description
1	N.C.
2	N.C.
3	TX A, To the network
4	RX A, From the network
5	RX B, From the network
6	TX , To the network
7	N.C.
8	N.C.

U0 NETWORK:

Pin	Description
1	N.C.
2	N.C.
3	N.C.
4	RING
5	TIP
6	N.C.
7	N.C.
8	N.C.

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14.5. External DC connector

Pin 1 : Minus
Pin 4 : Plus

14.6. Environment

Operating temp. Range:	0°C to 45°C (41°F to 113°F)
Humidity:	0 to 90% non -condensing
Storage temp. :	- 20°C to 60°C (-4°F to 140°F)
Dimensions:	(D x W x H) 270 x 482 x 45 mm
Weight:	**3.43** kg,

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